AMENDMENTS TO THE CLAIMS

The following Listing of Claims, with amendments to claims 1, and 8, will replace all prior versions, and listings, of claims in the application. Note that claims 22-50 were withdrawn in a prior amendment and remain withdrawn at this time.

1 (Currently Amended). A system for providing adaptive playback of an audio signal received across a packet based network, comprising:

storing data packets comprising a received audio data signal to a signal buffer;

outputting parts of the signal present in the signal buffer as needed for signal playback;

analyzing the data packets contained in the signal buffer to determine whether any data packets are missing, having not been received into the signal buffer by an expected arrival time<u>, said expected arrival time representing a predetermined packet late loss time</u>;

~~determining~~ <u>specifying</u> a maximum delay period<u>, extending past the expiration of the expected arrival time,</u> for receiving any missing data packets ~~based on a current level of the signal buffer~~;

<u>following the expiration of the expected arrival time,</u> stretching at least part of the signal preceding the missing data packets present in the signal buffer, until any of receiving the missing data packets and exceeding the maximum delay period, when the analysis of the contents of the signal buffer indicates that the length of the signal in the signal buffer is less than a predetermined threshold; and

compressing at least part of the signal present in the signal buffer when the analysis of the contents of the signal buffer indicates that the length of the signal in the signal buffer is greater than a predetermined threshold.

2 (Original). The system of claim 1 wherein analyzing the contents of the signal buffer includes determining a type of the contents of the signal buffer from among a group including: periodic content, quasi-periodic content, aperiodic content and mixed content.

3 (Original). The system of claim 2 wherein stretching at least part of the signal having any of periodic content and quasi-periodic content type comprises:

identifying at least one of the segment of the content of the signal buffer as a template;

searching for a matching segment in portions of the content of the signal buffer whose cross correlation peak exceeds a predetermined threshold; and

inserting the template into the content of the signal buffer, and aligning and merging the matching segments.

4 (Original). The system of claim 2 wherein stretching at least part of the signal having aperiodic content the type comprises automatically generating and inserting at least one synthetic segment into the buffered signal to increase the length of the content of the signal buffer.

5 (Original). The system of claim 4 wherein automatically generating the at least one synthetic segment comprises:

automatically computing the FFT of the at least part of the signal;

introducing a random rotation of the phase into the FFT coefficients; and

computing the inverse FFT for each segment, thereby creating the at least one synthetic segment.

6 (Original). The system of claim 4 wherein automatically generating the at least one synthetic segment comprises:

applying at least one LPC filter to the at least part of the signal to compute an LPC residual;

computing at least one FFT from the LPC residual;

introducing a random rotation of the phase into the coefficients of at least one of the computed FFTs;

computing inverse FFTs from the FFT coefficients to reconstruct the LPC residual; and

applying at least one inverse LPC filter to the LPC residual, thereby creating the at least one synthetic segment.

7 (Original). The system of claim 1 wherein the predetermined threshold for stretching and compressing at least part of the signal present in the signal buffer are optimized to compensate for clock drift between an encoder and a decoder.

8 (Currently Amended). A system for providing an adaptive playback of received frames of an audio signal transmitted across a packet-based network, comprising:

receiving and decoding data frames of an audio signal transmitted across a packet-based network;

storing the decoded data frames to a signal buffer;

analyzing the contents of the signal buffer to determine whether any data frames are missing due to corresponding data packets having not been received by an expected arrival time, said expected arrival time representing a predetermined packet late loss time;

~~determining~~ specifying a maximum delay period, extending past the expiration of the expected arrival time, for receiving any missing data packets ~~based on a current level of the signal buffer~~;

outputting one or more of the decoded frames present in the signal buffer when the analysis of the contents of the signal buffer indicates that the length of the signal in the signal buffer is between a predetermined minimum and a predetermined maximum buffer size;

following the expiration of the expected arrival time, stretching and outputting one or more decoded frames preceding the missing data packets in the signal buffer, until any of receiving the missing data packets and exceeding the maximum delay period, when the analysis of the contents of the signal buffer indicates that the length of the decoded frames in the signal buffer is less than the predetermined minimum buffer size; and

compressing and outputting one or more decoded frames in the signal buffer when the analysis of the contents of the signal buffer indicates that the length of the decoded frames in the signal buffer is greater than the predetermined maximum buffer size.

9 (Original). The system of claim 8 wherein any frame output from the signal buffer is removed from the signal buffer as it is output.

10 (Original). The system of claim 8 further comprising packet loss concealment for signal packets declared to be late loss packets.

11 (Original). The system of claim 8 wherein stretching and outputting one or more decoded frames provides automatic jitter control as a function of buffer content.

12 (Original). The system of claim 11 wherein stretching one or more decoded frames further comprises automatically determining a content type of the stretched frames prior to stretching those frames.

13 (Original). The system of claim 12 wherein the content type includes any of voiced framed, unvoiced frames, and mixed frames.

14 (Original). The system of claim 13 wherein stretching any voiced frame comprises:

identifying at least one of the segment of the voiced frame as a template;

searching for a matching segment in adjacent frames whose cross correlation peak exceeds a predetermined threshold; and

aligning and merging the matching segments of the frame.

15 (Original). The system of claim 8 wherein stretching any unvoiced frame comprises automatically generating and inserting at least one synthetic segment into the current frame to increase a length of the current frame.

16 (Original). The system of claim 15 wherein automatically generating the at least one synthetic segment comprises:

automatically computing the FFT of the current frame;

introducing a random rotation of the phase into the FFT coefficients; and

computing the inverse FFT for each segment, thereby creating the at least one synthetic segment.

17 (Original). The system of claim 15 wherein automatically generating the at least one synthetic segment comprises:

applying at least one LPC filter to the current frame to compute an LPC residual;

computing at least one FFT from the LPC residual;

introducing a random rotation of the phase into the coefficients of at least one of the computed FFTs;

computing inverse FFTs from the FFT coefficients to reconstruct the LPC residual; and

applying at least one inverse LPC filter to the LPC residual, thereby creating the at least one synthetic segment.

18 (Original). The system of claim 8 wherein stretching any mixed frame comprises:

identifying at least one segment of the frame as a template;

searching for a matching segment whose cross correlation peak exceeds a predetermined threshold;

aligning and merging the matching segments of the frame to create an interim voiced segment;

automatically generating and inserting at least one synthetic segment into the current frame to create an interim unvoiced segment;

weighting each of the interim voiced segment and the interim unvoiced segment relative to a normalized cross correlation peak computed for the current segment; and

adding and windowing the interim voiced segment and the interim unvoiced segment to create a partially synthetic stretched segment.

19 (Original). The system of claim 8 wherein compressing any voiced frame comprises:

identifying at least one segment of the frame as a template;

searching for a matching segment whose cross correlation peak exceeds a predetermined threshold;

cutting out the signal between the template and the match; and

aligning and merging the matching segments of the frame.

20 (Original). The system of claim 8 wherein compressing any voiced frame comprises:

shifting a segment of the frame from a first position in the frame to a second position in the frame;

deleting the portion of the frame between the first position and the second position; and

adding the shifted segment of the frame to the signal representing the remainder of the frame by using a sine windowing function for blending the edges of the segment with the signal representing the remainder of the frame.

21. The system of claim 8 wherein both the predetermined minimum buffer size for stretching one or more decoded frames in the signal buffer and the predetermined maximum buffer size for compressing one or more decoded frames in the signal buffer are optimized to compensate for clock drift between an encoder and a decoder.

22 (Withdrawn). A method for adaptive playback of received frames of an audio signal transmitted across a packet-based network, comprising using a computing device to:

receive a packetized audio signal broadcast across a packet-based network;

decode each received packet and store the resulting decoded signal frame in a signal buffer;

output a current packet in the case where the current packet has been received across the packet-based network;

instantiate a mute mode whereby a playback of the audio signal is at least partially muted when a maximum delay time for receiving the current packet has been exceeded, and the current packet has not been received;

instantiate a packet loss concealment mode whereby the playback of the audio signal is modified for reducing audible artifacts resulting from one or more lost packets when a current buffer content has been previously temporally stretched, the current packet has not yet been received, and a packet subsequent to the current packet has already been received.

23 (Withdrawn). The method of claim 22 further comprising analyzing the content of the signal buffer for determining a current length of the contents of the signal buffer.

24 (Withdrawn). The method of claim 23 further comprising stretching and outputting one or more decoded frames from the signal buffer when the current length of the contents of the signal buffer is less than a predetermined minimum buffer size.

25 (Withdrawn). The method of claim 24 wherein the predetermined minimum buffer size is optimized to compensate for clock drift between an encoder and a decoder.

26 (Withdrawn). The method of claim 23 further comprising compressing and outputting one or more decoded frames from the signal buffer when the current length of the contents of the signal buffer is greater than a predetermined maximum buffer size.

27 (Withdrawn). The method of claim 24 wherein the predetermined maximum buffer size is optimized to compensate for clock drift between an encoder and a decoder.

28 (Withdrawn). The method of claim 22 wherein modification of the playback of the audio signal is in the packet loss concealment mode comprises:

computing an average energy for a frame in the signal buffer immediately preceding the current packet that has not yet been received;

computing an average energy for a frame in the signal buffer immediately succeeding the current packet that has not yet been received; and

determining a target frame size for both the preceding and succeeding frames as a function of the ratio of the of the average energy of the succeeding frame to the preceding frame.

29 (Withdrawn). The method of claim 28 wherein determining a target frame size for both the preceding and succeeding frames further comprises stretching the succeeding frame and the preceding frames by an amount that is inversely proportional to the ratio of the average energy.

30 (Withdrawn). The method of claim 29 wherein instantiating the mute mode comprises generating and providing playback of a comfort noise signal to replace lost packets, said comfort noise signal being generated from at least one signal frame stored in a silence buffer, said signal frame having been determined to represent nominal background noise.

31 (Withdrawn). The method of claim 30 further comprising periodically replacing the signal frames in the silence buffer as a function of a computed energy of those frames.

32 (Withdrawn). The method of claim 30 wherein generating the comfort noise signal from the at least one signal frame stored in a silence buffer comprises:

automatically computing the FFT of the at least one signal frame stored in the silence buffer;

introducing a random rotation of the phase into the FFT coefficients;

computing the inverse FFT for each segment, thereby creating the at least one synthetic silence segment; and

providing the at least one silence segment for playback as the comfort noise signal.

33 (Withdrawn). A computer-readable medium having computer executable instructions for providing adaptive decoding and playback of a packetized audio signal, said computer executable instructions comprising:

receiving a plurality of network packets, said network packets representing a packetized audio signal;

decoding each network packet as it is received and storing the decoded packet as a signal frame in a signal buffer;

estimating an LPC filter for each signal frame, computing an LPC residual from each signal frame using the estimated LPC filter, and storing each LPC residual in an LPC residual buffer;

examining a current length of the LPC residual buffer;

stretching and outputting a current LPC residual from the LPC residual buffer when the current length of the LPC residual buffer is less than a predetermined minimum buffer size; and

computing an inverse LPC of the stretched LPC residual, and outputting the result as a current signal frame.

34 (Withdrawn). The computer-readable medium of claim 33 wherein the predetermined minimum buffer size is optimized to compensate for clock drift between an encoder and a decoder.

35 (Withdrawn). The computer-readable medium of claim 33 further comprising:

compressing and outputting a current LPC residual from the LPC residual buffer when the current length of the LPC residual buffer is greater than a predetermined maximum buffer size; and

computing an inverse LPC of the compressed LPC residual, and outputting the result as a current signal frame.

36 (Withdrawn). The computer-readable medium of claim 35 wherein the predetermined maximum buffer size is optimized to compensate for clock drift between an encoder and a decoder.

37 (Withdrawn). The computer-readable medium of claim 33 further comprising instantiating a mute mode whereby a playback of the audio signal is at least partially

muted in the case where a maximum delay time for receiving a current packet has been exceeded, and the current packet has not been received.

38 (Withdrawn). The computer-readable medium of claim 33 further comprising instantiating a packet loss concealment mode whereby a playback of the audio signal is modified for reducing audible artifacts resulting from one or more lost packets in the case where a current LPC residual buffer content has been previously stretched, a current packet has not yet been received, and a packet subsequent to the current packet has already been received.

39 (Withdrawn). A method for providing adaptive signal playback, comprising using a computing device to:

receive signal packets representing a digitized audio signal transmitted across a packet-based network;

decode the packets to reconstruct the digitized audio signal;

store the reconstructed digitized audio signal in a signal buffer;

provide content of the signal buffer for playback as required by a playback device;

begin stretching contents of the signal buffer when an expected signal packet has not been received at an expected time; and

continue stretching contents of the signal buffer until a condition selected from (1) actual receipt of the expected signal packet, and (2) a determination that the expected signal packet is lost.

40 (Withdrawn). The method of claim 39 wherein the determination that the expected signal packet is lost is a function of the amount of stretching already applied to the contents of the signal buffer, receipt of one or more subsequent expected signal packets, and existing content of the signal buffer.

41 (Withdrawn). The method of claim 39 further comprising muting playback of the audio signal when a predetermined delay time has been exceeded without receiving any signal packets.

42 (Withdrawn). The method of claim 39 further comprising stretching contents of the signal buffer when the length of the contents in the signal buffer is less than a predetermined threshold.

43 (Withdrawn). The method of claim 42 wherein the predetermined threshold is optimized to compensate for clock drift between an encoder and a decoder.

44 (Withdrawn). The method of claim 39 further comprising compressing contents of the signal buffer when the length of the contents in the signal buffer exceeds a predetermined threshold.

45 (Withdrawn). The method of claim 44 wherein the predetermined threshold is optimized to compensate for clock drift between an encoder and a decoder.

46 (Withdrawn). The method of claim 39 further comprising removing content from the signal buffer as it is provided for playback as required by a playback device.

47 (Withdrawn). The method of claim 39 further comprises analyzing contents of the signal buffer to determine a content type of at least part of the contents of the signal buffer.

48 (Withdrawn). The method of claim 39 wherein the content type is quasi-periodic, and wherein stretching contents of the signal buffer comprises:

identifying at least one of the segment of the voiced frame as a template;

searching for a matching segment in adjacent frames whose cross correlation peak exceeds a predetermined threshold; and

aligning and merging the matching segments of the frame.

49 (Withdrawn). The method of claim 39 wherein the content type is aperiodic, and wherein stretching contents of the signal buffer comprises:

computing at least one FFT from at least one part of the contents of the signal buffer;

randomizing a phase rotation of the coefficients of at least one of the computed FFTs;

computing an inverse FFT from the coefficients for each FFT to synthesize a signal segment corresponding to each computed FFT; and

stretch at least part of the contents of the signal buffer by inserting each synthesized signal segment into the buffered audio signal.

50 (Withdrawn). The method of claim 49 further comprising:

applying an estimated LPC filter to the contents of the signal buffer to compute an LPC residual for use in place of the contents of the signal buffer for computing the at least one FFT from at least one part of the contents of the signal buffer; and

applying an interpolated inverse LPC filter to the signal segment corresponding to each computed FFT prior to stretching at least part of the contents of the signal buffer by inserting each synthesized signal segment into the buffered audio signal.